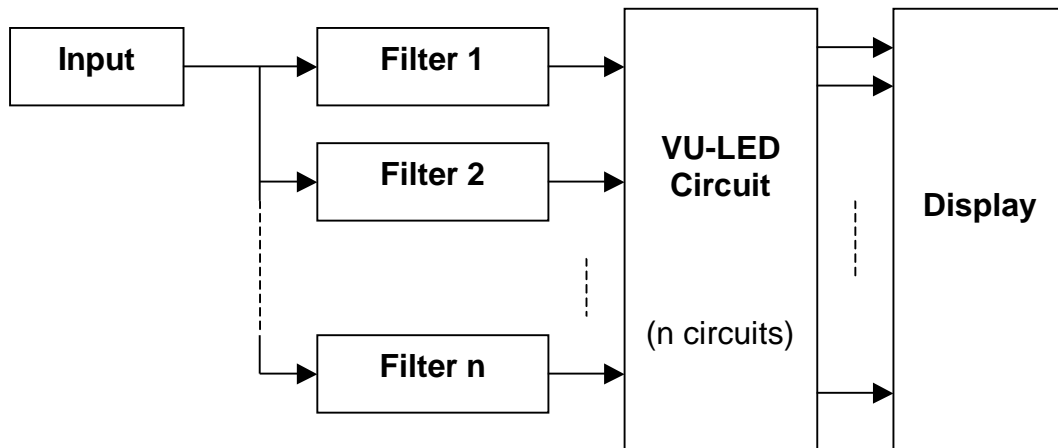


PART I: INTRODUCE BLOCK DIAGRAM AND OPERATION THEORY

I. Basic block diagram of audio spectrum analyzer:

1. Block diagram:



Basic block diagram of audio spectrum analyzer

2. Operation theory:

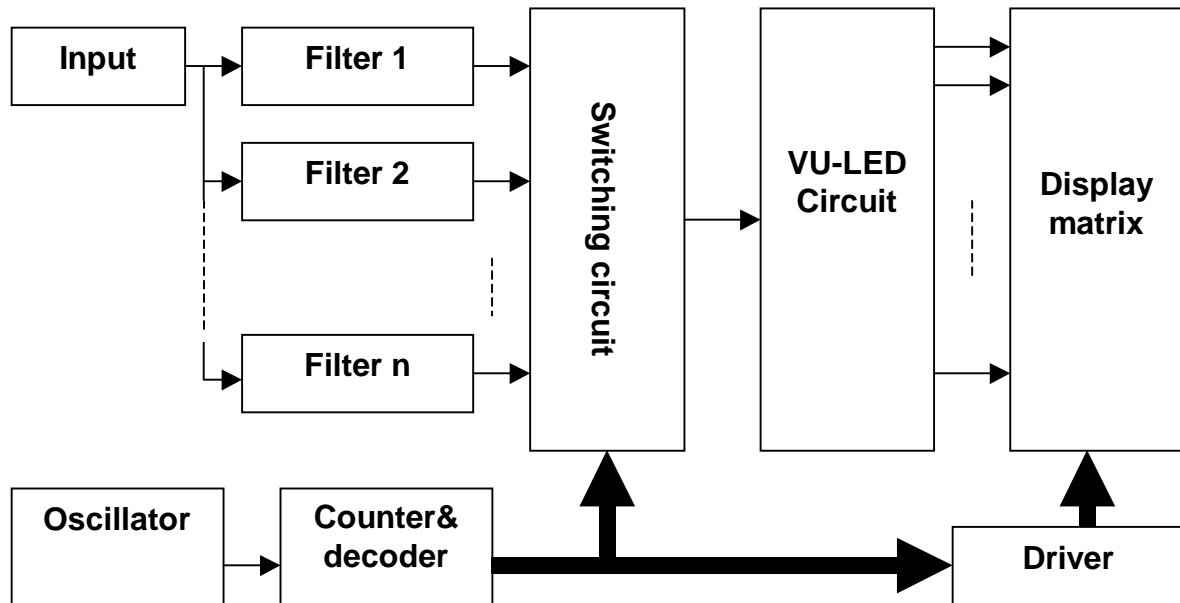
Input signal comes into each filters. Filters are band pass filters. They only pass signals belonged to fixed frequency range and remove others. We can use transistor and discrete capacitors and resistors to make active filters or we can use OP-AMP and other passive device. The more filters, the higher frequency resolution of spectrum analyzer, shown as the higher quality of the circuit.

VU-LED Circuit employed to display signal level behind filters. This kind of display circuit can be discrete devices or specific ICs. The more outputs, the higher resolution of amplitude.

In the above diagram, we know that each filter needs and VU-LED circuit for display, the more filters (the higher frequency resolution of the circuit), the more VU-LED circuits. This matter will make the circuit complex for wiring. To overcome this weak point, we will consider following improvement in those block diagrams:

II. Block diagram of audio spectrum analyser uses multiplex display:

1. Block diagram:



Block diagram of audio spectrum analyser uses multiplex display

2. Operation theory:

The advantage of this diagram is only one VU-LED circuit employed, regardless the number of filters, the number of wiring to display circuit is also more simple because display matrix is the combination of rows & columns. The number of columns is adapted to the number of filters, and the number of rows is the number of VU-LED outputs.

This diagram also includes n filters, outputs of the filters are connected to switching circuit, at one time, switch passes signal from only one filter. Switch is controlled by counter & decoder, clocking signal from oscillator comes to counter & decoder, the number of outputs of counter & decoder is n . Outputs of counter & decoder are also connected to driving circuit to scan columns. If the frequency of clock is fast enough then our eyes are cheated to have the feeling that all columns are bright simultaneously but actually, at one time, only one column is bright.

PART II: FILTERS

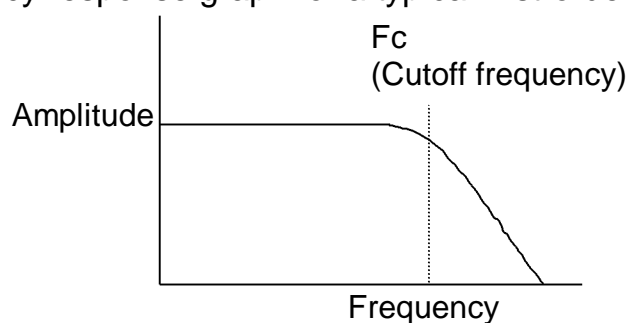
I. Introduce filters:

Generally, in electronic equipment, if they need to reject or pass any frequencies they often use frequency filters. At first, it's mainly constructed by inductor L and capacitor C. Nowadays, OP AMP with small dimension, many characteristics, low cost, simple in design calculation so it's preferred to make active filter RC.

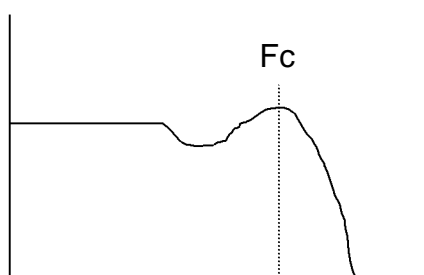
There are many kinds of filters such as Butterworth and Chebyshev. Before considering filters, we must consider filter orders. The order of a filter identifies its cutoff slope. The higher the order number, the steeper the cutoff slope is. Filter orders increase in steps of 6dB/octave. The simplest filter is a first-order filter has a cutoff slope of 6dB/oct. Other high-grade active filters may have higher orders, for example, a second-order filter has a cutoff slope of 12dB/oct. The following part, as per project requirements, mainly introduces about bandpass filters. Following is an introduction about filters:

1. Low pass filters:

A Butterworth filter is designed to have a very flat frequency response within its bandpass and a smooth, uniform roll-off characteristic. Below figure shows a frequency response graph for a typical first-order Butterworth filter.



Another commonly used is the Chebyshev filter. The frequency response graph of a Chebyshev low-pass filter is shown in the figure below.

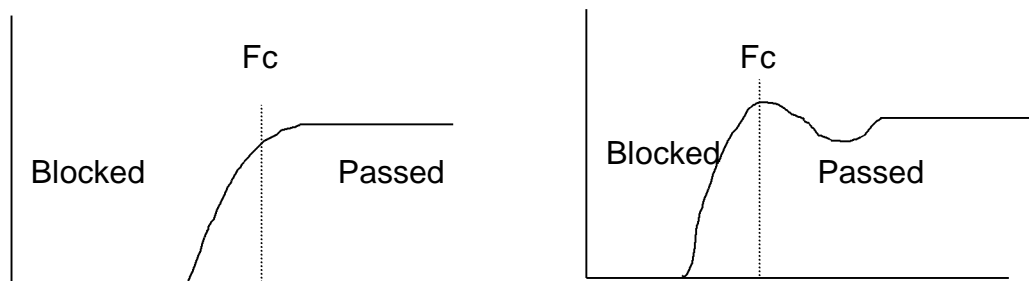


Notice that the frequency response is not flat smooth below the cutoff frequency as was the case with the Butterworth filter. In the Chebyshev filter, there is a slight dip in amplitude below the cutoff frequency then the amplitude goes back up to the flat level just before the actual cutoff slope begins. The chief advantage of the Chebyshev filter is its very steep roll-off characteristic.

Actual circuits for Butterworth filter and Chebyshev filter are usually quite similar. Often, the only real difference in the two filter types is in the actual component values used.

2. High pass filters:

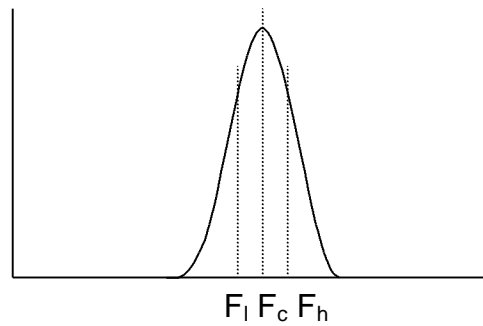
Functionally, a high pass filter is the exact opposite of a low pass filter. Ignoring the roll-off slope, whatever is passed by the lowpass filter is blocked by the highpass filter and vice versa. Active highpass filter circuits are remarkably similar to active lowpass filters, except the position of some components are changed. Like the active lowpass filter, an active highpass filter can have either a Butterworth response or a Chebyshev response.



3. Band pass filters:

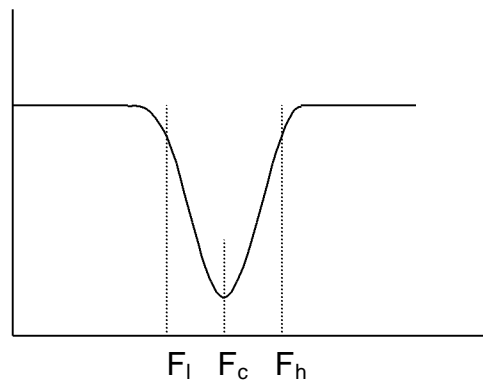
Generally speaking, active bandpass filters are more complex than active lowpass or highpass filters. In a sense, lowpass and highpass filters are bandpass filters of a sort. In a lowpass filter, the lower cutoff frequency is at some imaginary point below 0Hz. For a highpass filter, the upper end of the passband is determined by the frequency response of the OP-AMP (or other active device) used to build the filter circuit.

A bandpass filter can be created by placing a lowpass filter and a highpass filter in series. Bandpass filters are more complex because they control more variables so they are more versatile. Variables include: gain (K), filter order (n), center frequency (F_c), and bandwidth (BW). Moreover, another variable is quality factor Q , derived from F_c and BW .



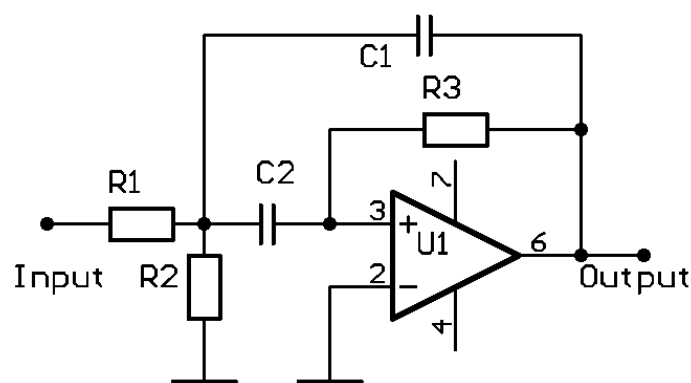
4. Band-reject filters:

A band-reject filter passes almost frequencies except frequencies belonged to determined range (often narrow). Band-reject filters are often used to remove unwanted frequencies. Higher frequencies and lower frequencies than blocking range will be passed easily. In frequency response graph, we can see an hole or dip so this circuit also called Notch filter.



II. Details about bandpass filter:

Basic bandpass filter as illustration below:



Schematic diagram of a basic bandpass filter

This circuit can be designed to have gain from low to medium and Q value can be up to 20, lower Q value can be obtained by choosing suitable component values.

In circuit, choose $C_1 = C_2 = C$ for easy calculation. The predefined parameters are center frequency(F_c), gain (K) and Q . In most of cases, Q value can be derived from center frequency and bandwidth of the bandpass filter.

$$Q = \frac{F_c}{BW}$$

With predefined parameters, we have formulas to calculate R_1 , R_2 and R_3 :

$$R_1 = \frac{Q}{(2pF_c CK)}$$

$$R_2 = \frac{Q}{(2pF_c C(2Q - K))}$$

$$R_3 = \frac{2Q}{(2pF_c C)}$$

Gain is determined by ratio of R_1 and R_3 :

$$K = \frac{R_3}{2R_1}$$

One important limit for this circuit is if high gain (K) then Q must be high. Cannot design filter circuit with high gain and low Q because value of R_2 will be negative. In the circuit, the order is determined by Q value. The higher Q value, the steeper roll-off is.

With predefined C , R_1 , R_2 , R_3 , we can calculate:

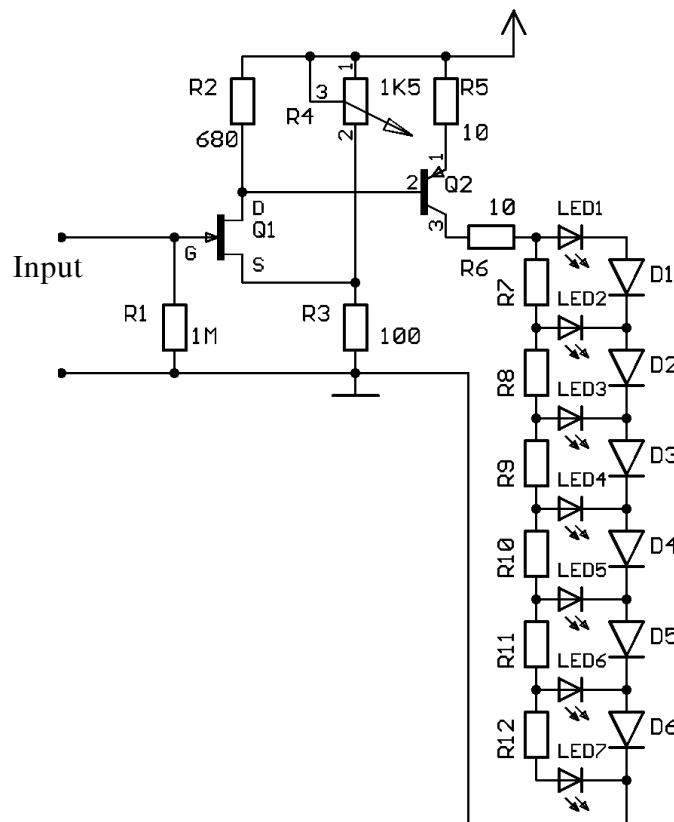
$$F_c = \frac{1}{2pC} \sqrt{\frac{R_1 + R_2}{R_1 \cdot R_2 \cdot R_3}}$$

$$Q = \pi F_c \cdot C \cdot R_3.$$

PART III: VU-LED CIRCUITS

I. VU-LED circuits use discretes:

1. Circuit employed transistors:



VU-LED circuit uses transistors

In above schematic, Q_1 and Q_2 create a two stages amplifier. When input has no signal, Q_1 almost closes (this status is determined by varistor R_4), voltage loss on R_2 is small, not enough to open Q_2 so on C pole of Q_2 there is no output current, all LEDs are off.

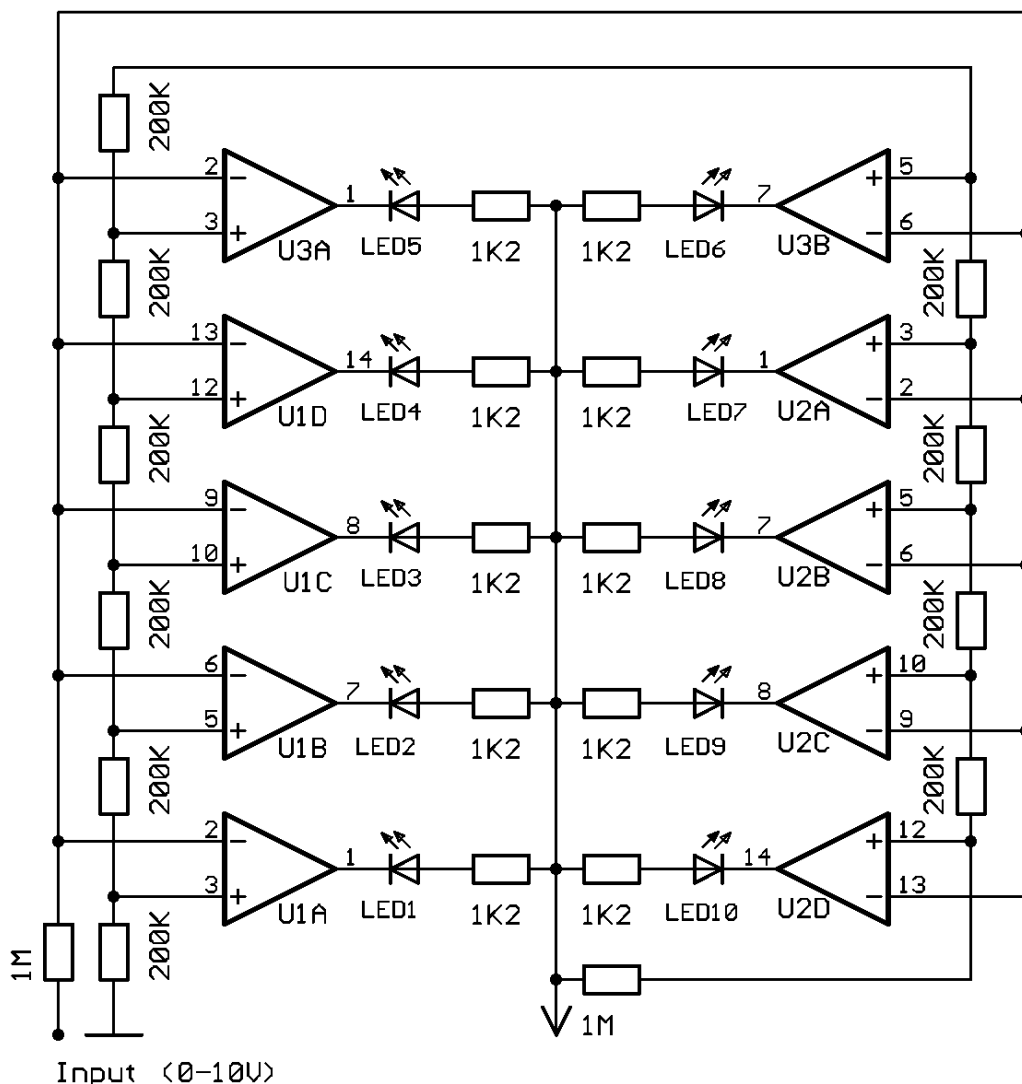
When a positive voltage is applied to input, Q_1 opens, the higher input voltage, the more Q_1 opens. Thus Q_2 also opens and there is output current on C pole. The stronger input voltage, the bigger output current. When output current appears, LEDs will be bright, from the last LED (LED₇)

When output current from C pole of Q_2 appears then this current almost goes through R_{12} and LED₇ so make voltage loss on this segment (at anod of LED₆ to mass). With a defined current, LED₇ is bright and voltage loss on it is about $1,8 \div 2V$. During increasing of current, this voltage is no changed.

Another hand, LED₇ functions as a voltage regulator. But increasing current will make increasing voltage at anod of LED₆. When this voltage reaches the value of total voltage loss on LED₇ and open diode D₆ (0,7V) that means about 2,5 ÷ 2,7V then LED₆ will be bright. LED₅ will be bright next when current from C pole of Q₂ continue increasing, when voltage at anod of LED₅ reaches value of total voltage loss on brighting LEDs and open diodes D₅, D₆. Generally, next LED only bright when increase voltage on its anod (compare to mass) up to 0,7V compare to voltage on anod of previous LED. When output current on C pole of Q₂ decreases, LEDs will be off by the order from top to bottom.

The linearity of indicating LED depend on choosing exact resistors R₇ ÷ R₁₂ and the same parameters of LEDs. This circuit not only works with constant signal source but also works with audio signal. In this case, the circuit works only with possitive half cycle of signal.

2. Circuit employed OP-AMP:



Voltage indicator circuit

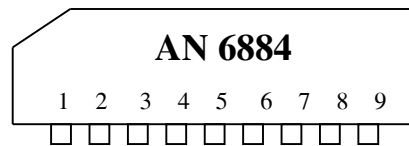
In this circuit, non-inverted inputs connect to bias circuit to have sample voltage, while voltage signal comes to inverted inputs. The circuits will compare the levels and make relative LEDs bright.

II. Introduce specific VU-LED chips:

1. AN 6884:

AN 6884 is a VU-LED IC with 9 pins, displays in bar mode. All pins are on one side. This IC has 5 outputs, output current is constant. Power supply $V_{cc} = 3,5 \div 16V$, maximum power dissipation $P_{Dmax} = 1100mW$, current supply $I_{cc} = 18mA$, output current $I_o = 15mA$, working temperature $T_{opr} = -20 \div 75^{\circ}C$. Direct input signal can be DC or AC.

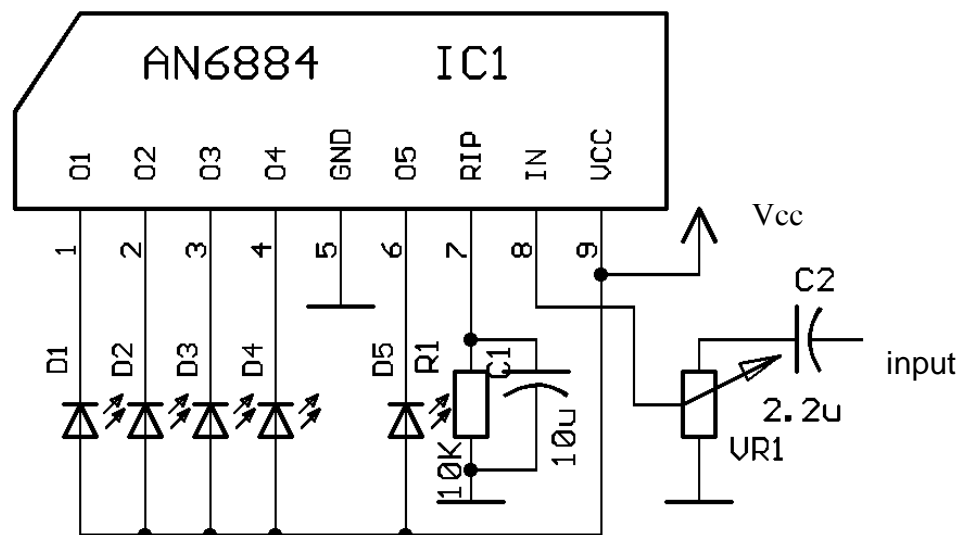
Pin assignment of AN 6884:



Function for each pins as following:

- Pin 1, 2, 3, 4, 6 are outputs.
- Pin 5 connects to mass, pin 9 connects to power supply +Vcc.
- Pin 8 is input.
- Pin 7 connects to lowpass filter R and C.

Application circuit of AN 6884:



Schematic of VU LED AN6884

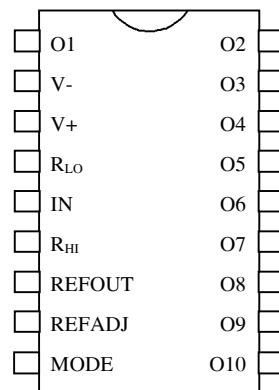
2. LM 3914:

LM 3914 is monolithic, it can drive 10 LEDs following analog input signal. The display is linear and it has one pin to select display mode is dot or bar.

Specification: package DIL 18 pins, total power dissipation $P_{Dmax} = 1365mW$ with maximum temperature $100^{\circ}C$, working voltage range $V_{cc} = 3 \div 18V$.

LM 3914 is used versatilely, outputs are current regulated and programmed so no need to use a traditional limit current resistor for LEDs. This feature allows power supply IC with low voltage down to 3V. LM 3914 has standard voltage source 1,25V so allows adjustment from 1,2 \div 12V and limit current for LEDs in range of 2 \div 30mA.

Pin assignment:



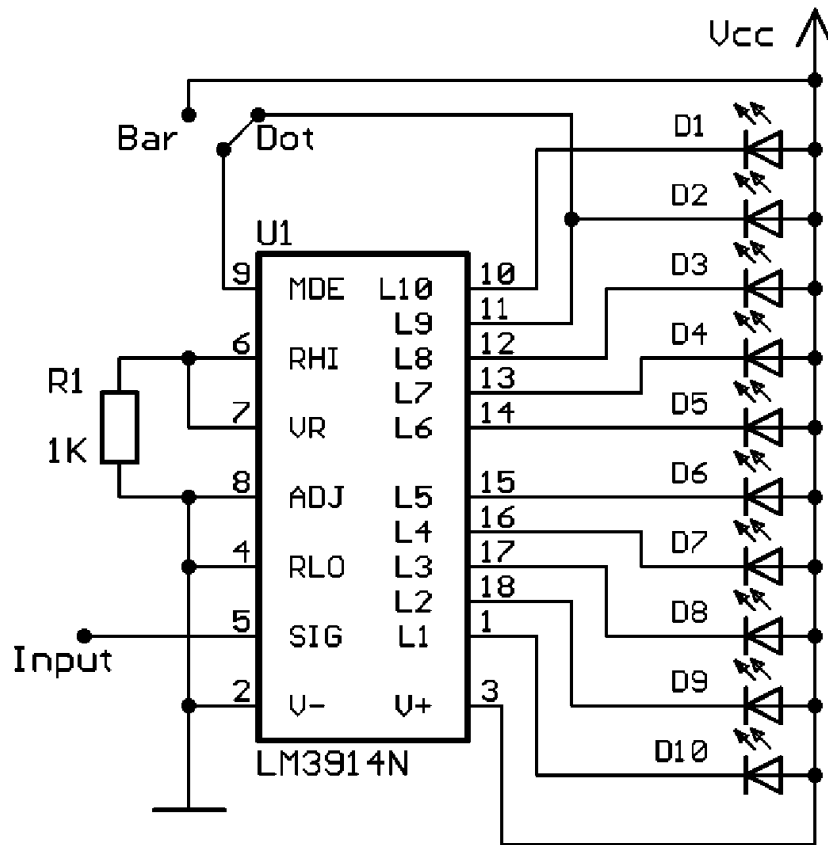
Functions of each pin:

- Pin 2, 3 : power supply V-, V+.
- Pin 1, 10-18: outputs.
- Pin 4, 6: outputs of voltage divide circuit.
- Pin 7: output of standard voltage.
- Pin 8: adjust standard voltage.
- Pin 9: select display mode. When connects to pin 11, LEDs display at dot mode, when connects to V+ then LEDs display at bar mode.

Available standard voltage at pin 7 often connects to 10 stage bias voltage divide circuit of non-inverted inputs of voltage comparators control outputs. Control voltage put into a buffer amplifier to protect over voltage and negative voltage by a resistor and a diode. The buffer has high input resistance, low bias current so IC can work with signals near zero level. Ten

comparators inside control one by one by buffers, this allows indicaton exact up to 0,5% in high temperature enviroment.

The following is application schematic of LM 3914:

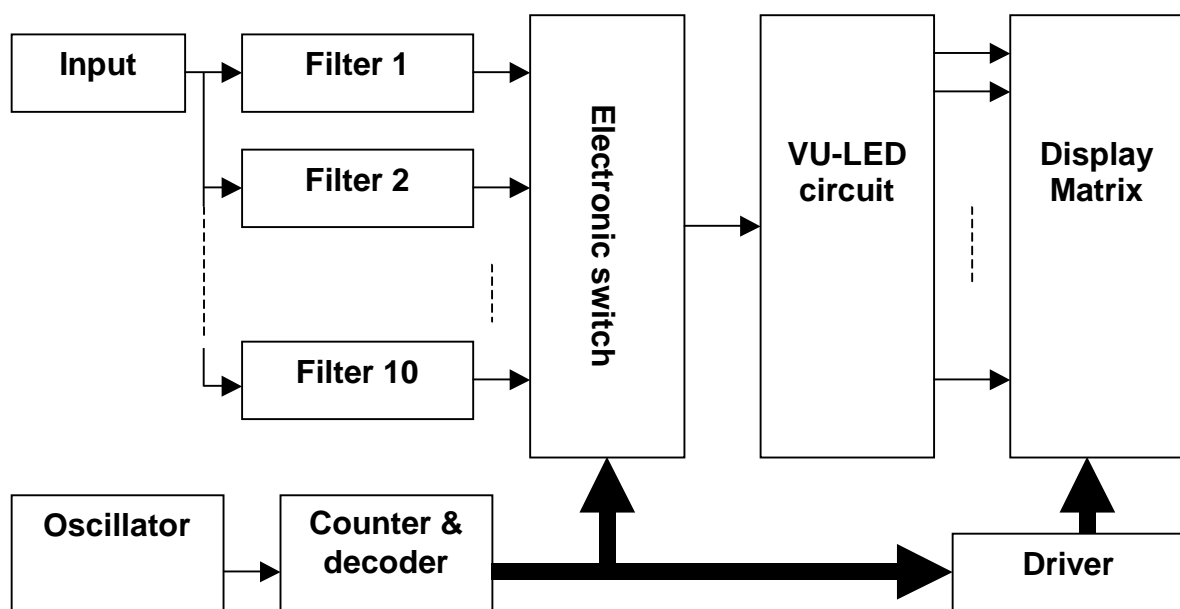


Application schematic diagram of LM3914

PART IV: DESIGN AUDIO SPECTRUM ANALYZER CIRCUIT

I. Block diagram, calculate parameters:

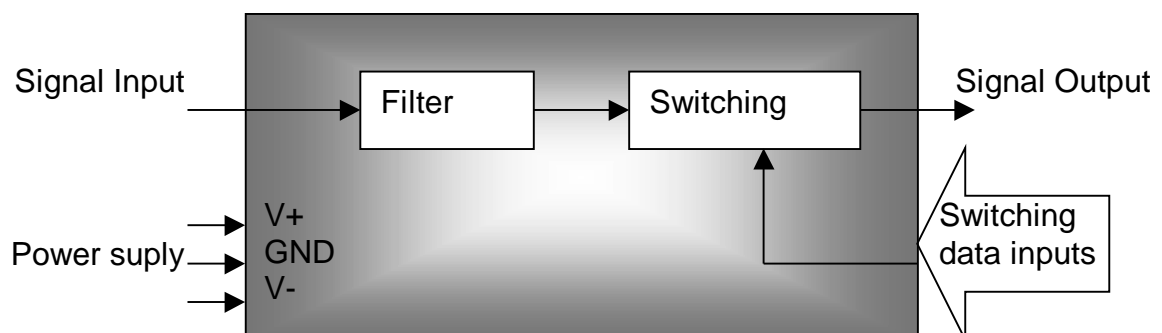
In this project, audio spectrum analyzer has resolution of frequency is 10, resolution of amplitude is also 10. The circuit uses multiplex display for small circuit, wiring is also little and reduces components.



Block diagram of audio spectrum analyzer circuit

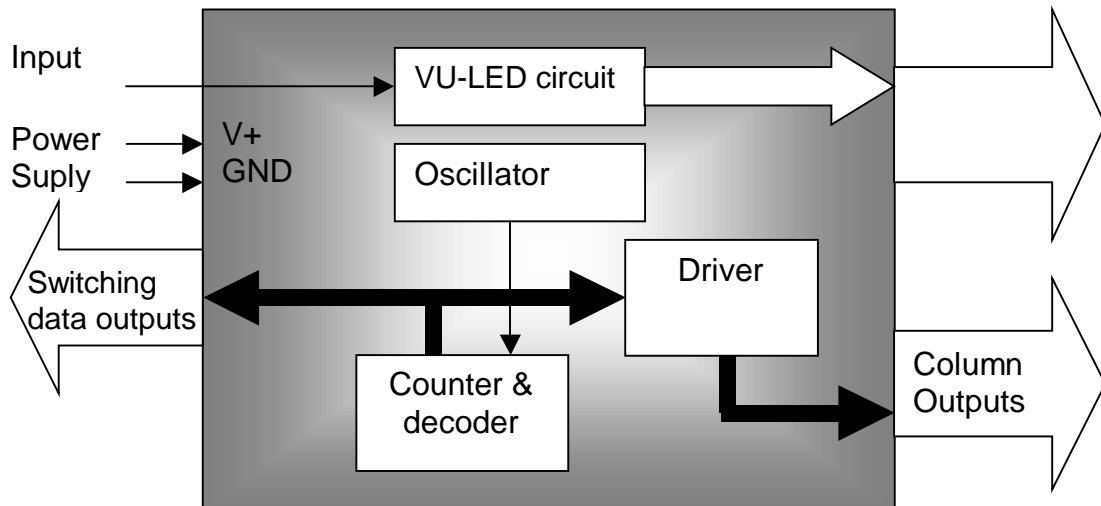
Devide the circuit into main parts: filter circuits block with electronic switchers; oscillator, counter, drivers block with VU LED circuit; and LED matrix display block.

1. Filters and switching block:



Signal from input is put simultaneously into all filters, signals from outputs of those filters are put into electronic switchers with input is switching data. At one time, there is only one output signal from one filter. This implemented by putting suitable switching data.

2. Oscillator, counter, driver with VU LED circuit:

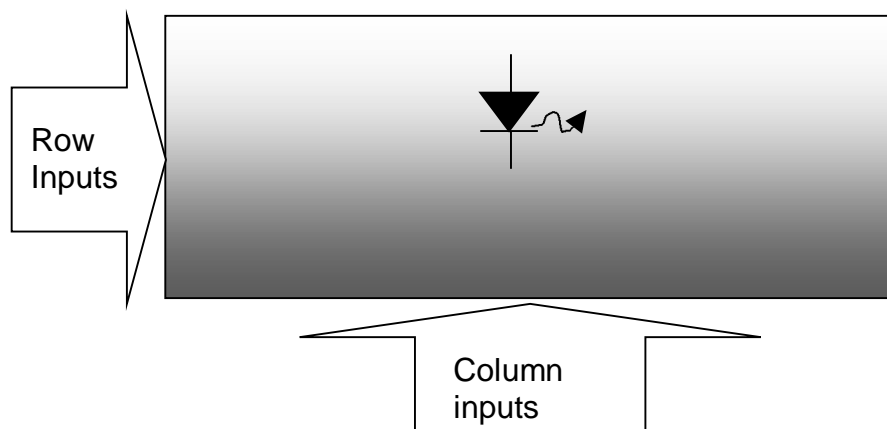


In this block, VU-LED is independent to other parts but for compact purpose so it's included into this part. Input signal is put into VU-LED circuit so we have relative row output data.

The oscillator makes clock pulse suplying to counter with decoder circuit, outputs of decoder are switching data outputs for switchs, simultaneously these outputs also are put into drivers to make outputs for columns sweeping.

3. LED display matrix block:

This block is the simplest. This is a matrix of rolws & columns. Cross point of row & column is LED connected. Data is put into row at relative time of selected column to display.



II. Design, calculation:

1. Filters and switching blocks:

a. Filters:

Depend on the quality, we can see audio spectrum analyzers are divided into center frequencies of filters such as 20Hz, 30Hz, 60Hz, 125Hz, 250Hz, 500Hz, 1KHz, 2KHz, 4KHz, 8KHz, 16KHz, 20KHz, In other simpler audio spectrum analyzers, we can see center frequencies of filters are only 5 frequencies, in some professional types, the number of center frequencies up to 10, 15, 16, 32,... and of course the higher the number of center frequencies the higher quality of the circuit is.

Generally, center frequencies are varied follow Octave rule (frequency double), that because of harmonic processing problem. This matter has strong influence to fidelity, quality of polyphonics such as a concert with many instruments. For example: usually we can distinguish Am note at the same Octave of a guitar and of a piano because harmonics at higher octaves of standard Am note which we can distinguish in two instruments. We can describe some frequencies which some instruments may obtain as follow:

- At frequency of 30Hz: Bass violin, Bass Tuba, Contrebass,...
- At frequency of 60Hz: Trombone, Bassoon, Cello,...
- At frequency of 100Hz: Viola, human voices, Kettle drum, guitar basses, drums,...
- At frequency of 330Hz: basic sound of instruments and human voices.
- At frequency of 1KHz: concentrate sound of instruments, high frequencies of human voices.
- At frequency of 3,3KHz: concentrate voice created by string instruments, very clear at neigbough of this frequency range.
- At frequency of 10KHz: concentrate high order harmonic of basic voice of instruments, noise of magnetic tape.
- At frequency over 10KHz: high order harmonic of basic voices and of some special instruments.

From above relative statistic, we have some ideas about công dụng của các bộ lọc. Filter designation can be divided into independent bandpass filters with specific center frequencies for calculation.

An active bandpass filter uses OP-AMP, we can use following formulas as result:

∅ Center frequency determined with capacitors $C_1 = C_2$.

∅ Resistors R_1, R_2 determine input impedance of the circuit: $Z = R_1 + R_2$.

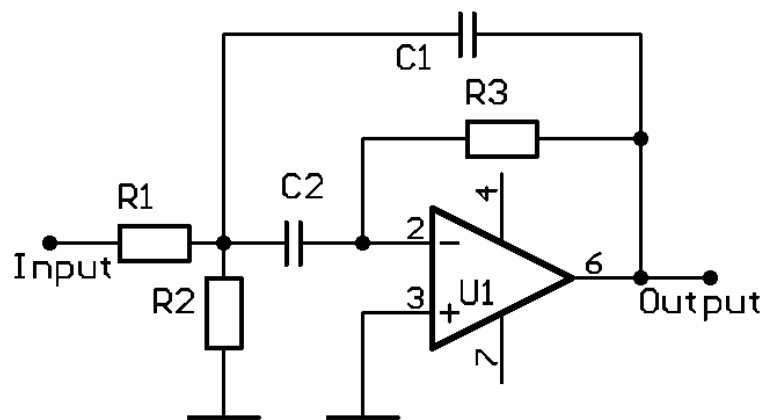
∅ Voltage gain of the circuit determined by: $K = \frac{R_3}{2R_1}$

∅ Quality factor: $Q = \pi F_c \cdot C \cdot R_3$.

∅ Center frequency determined by formula: $F_c = \frac{1}{2pC} \sqrt{\frac{R_1 + R_2}{R_1 \cdot R_2 \cdot R_3}}$

∅ Values R_1, R_2, R_3 are calculated by formulas:

$$R_1 = \frac{Q}{(2pF_c CK)} \quad R_2 = \frac{Q}{(2pF_c C(2Q - K))} \quad R_3 = \frac{2Q}{(2pF_c C)}$$



When calculating, we can predefine some parameters, after that derive remained values by above formulas.

In this design, choose $A_v = 12\text{dB}$ is the gain of the circuit as $K = 4$, quality factor $Q = 2$, resistors $R_1 = R_2 = 120\text{K}$, we can derive other parameters:

$$R_2 = 120\text{K}\Omega; \Rightarrow R_3 = 960\text{K}\Omega \text{ (choose } R_3 = 1\text{M}\Omega)$$

$$Q = \pi F_c \cdot C \cdot R_3$$

$$\Rightarrow F_c = \frac{Q}{pCR_3} = \frac{2}{3,14 \cdot 10^6 \cdot C}$$

$$\Rightarrow C = 0,636 \frac{10^{-6}}{F_c} (F)$$

To obtain different frequencies for filters, we can change F_c to calculate value $C_1 = C_2 = C$.

In this design, we choose 10 filters with center frequencies as follow: 32Hz, 64Hz, 125Hz, 250Hz, 500Hz, 1KHz, 2KHz, 4KHz, 8KHz, 16KHz. The calculated result as below table:

Fc (Hz)	C ₁ = C ₂ (theory)	C ₁ = C ₂ (actual)
32	19,87 nF	22 nF
64	9,93 nF	10 nF
125	5,08 nF	4,7 nF
250	2,54 nF	2,2 nF or 2,7 nF
500	1,27 nF	1,2 nF or 1,5 nF
1K	636 pF	620 pF
2K	318 pF	330 pF
4K	159 pF	160 pF
8K	79,5 pF	82 pF
16K	39,7 pF	39 pF

Input resistance of OP AMP is high so filters almost do not depend on OP AMP types (except for frequency response). Input resistance of this filter is $\frac{R_1 + R_2}{10}$ (because there are 10 separate filter circuits put in parallel).

OP AMPs in this circuit use dual power supply. Because we need 10 filters so we use 2 ICs with 4 OP AMPs inside and 1 IC with 2 OP AMPs. ICs used are TL084 (4 OP-AMPs) and TL082 (2 OP-AMPs).

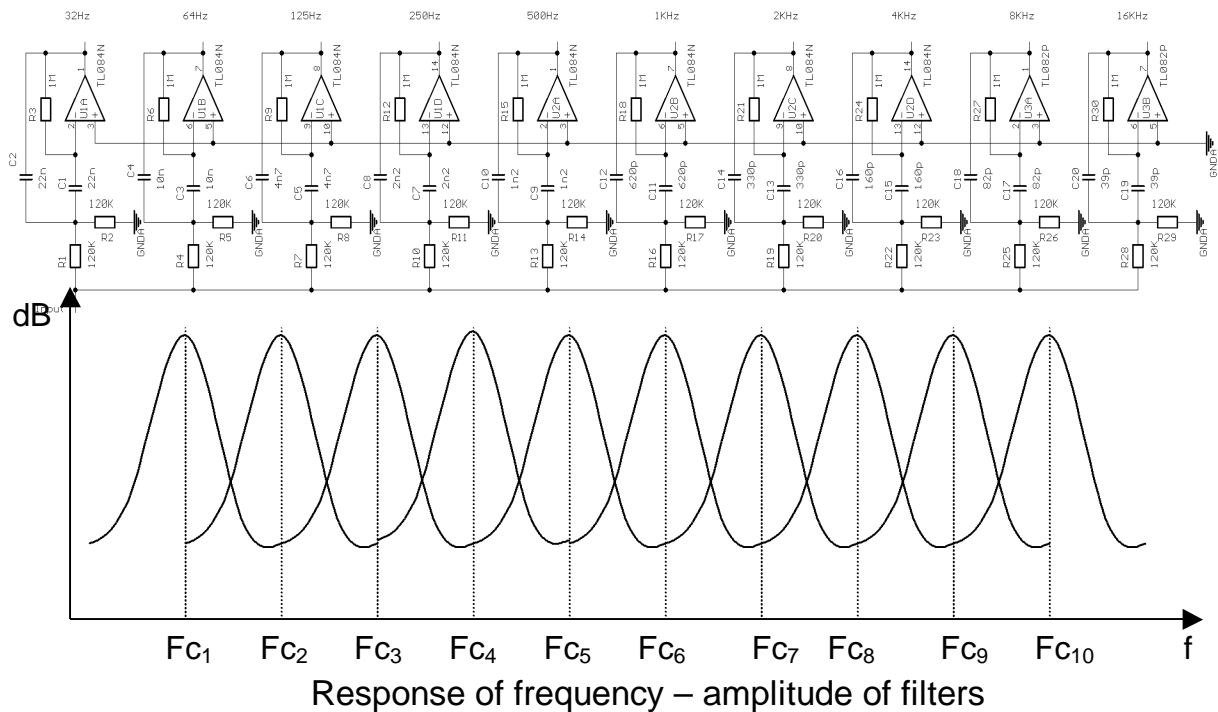
The following is specification of two ICs:

- Working voltage: $\pm 3 \div \pm 18V$.
- Offset voltage $V_{os}(\max)$: 15mV.
- Input bias $I_B(\max)$: 400pA.
- Bandwidth: 3MHz.
- Maximum input voltage: $\pm 30V$.
- Increasing voltage speed: 13V/ μs .
- Input resistance: $> 10^{12}\Omega$.
- Consumption current(max): 5,6mA for TL082 and 11,2mA for TL084.

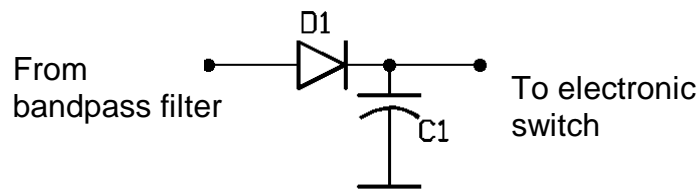
Specification at condition of $V_{cc} = \pm 15V$, temperature $T = 25^\circ C$.

- Open loop gain: 106dB.
- Input bias current: 30pA.
- Output amplitude $\pm 13,5V$.

We have complete 10 filter circuits as below:

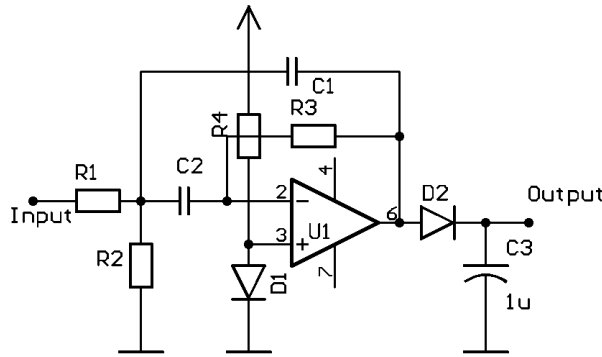


To have average voltage to put to electronics switches so at outputs of each filters we need rectifying circuit.



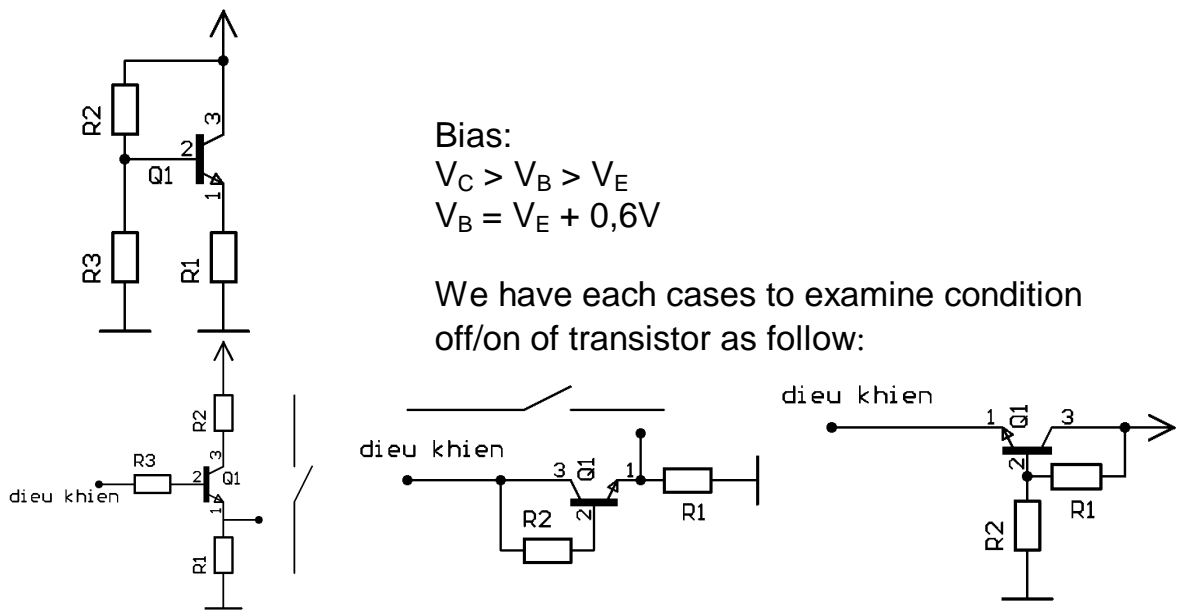
Capacitor C_1 can be chosen by practising to have the optimal display (up and down speed of display column must not be too fast or too slow compare to music, typically choose $C_1 = 1\mu F$).

Because DC level at filters' output is nearly 0V, so after rectifying circuit voltage will be lost 0,6V on rectifying diode. So filters' output must be raised DC level to 0,6V to compensate voltage loss after rectifying circuit. This matter is implemented as follow:

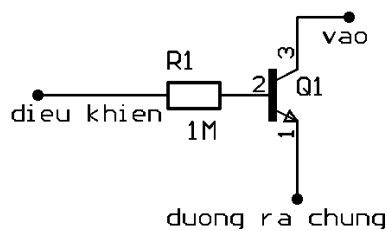


b. Electronic switches:

In electronic circuit, switches are used commonly. Switches can be diodes, transistors, ... This part only illustrates about switches use transistor.



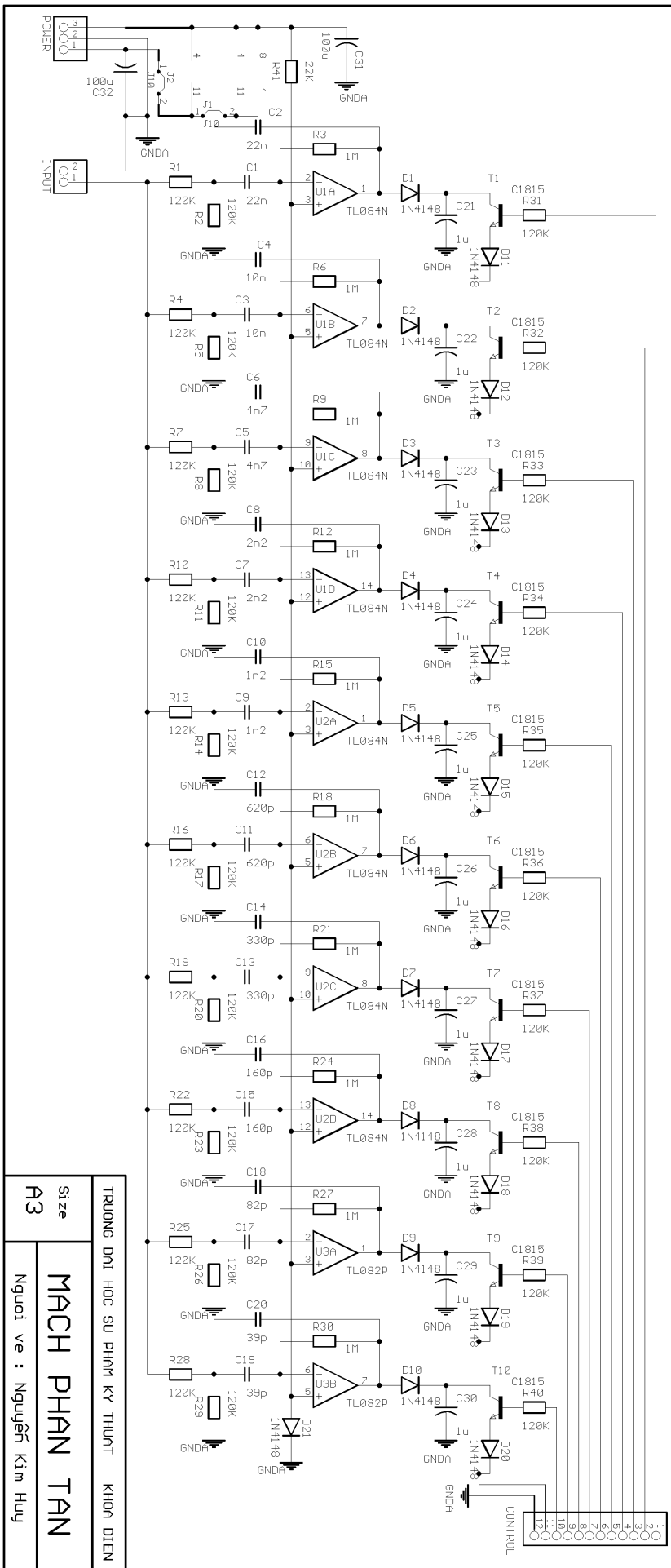
Above schematic is showed for output close or open to power suply (or to mass). Our need is separate input and output. We consider this schematic:



When control input is at high voltage level will make transistor open, make connection between input and output. resistor R must be chose to make input current I_B not to affect to output (I_E) when no signal at input. R is chose by practising. Q is low power transistor to have high β .

We have complete schematic diagram of filter and electronics switches as follow:

Schematic diagram of bandpass filters and electronic switches

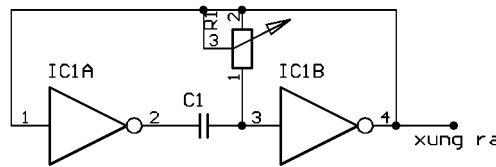


TRƯỜNG ĐẠI HỌC SƯ PHẠM KỸ THUẬT KHÓA ĐIỆN
Size A3
MẠCH PHÂN TÁN
Người vẽ : Nguyễn Kim Huy

2. Oscillator, counter & decoder, driver with VU LED:

a. Oscillator:

An oscillator makes clock pulse can be created from two inverting gates. This circuit as follow:



This circuit uses two inverter stage IC1A and IC1B create an astable circuit to make square pulse. Output from IC1B stage connects directly to input of IC1A. Capacitor C_1 make positive feedback between IC1A and IC1B. R_1 and C_1 are timing circuit. By this kind of circuit, output frequency can be calculate as this formula:

$$f = \frac{1}{1.4R_1C_1}$$

In the circuit, pulse clock controls sweeping speed of switches and display. In one cycle, only one frequency is compared amplitude and only one LED column is bright with height related to amplitude. If we make the LED column is bright equal or more than 25 times for 1 second so our eyes can not realize flashing of LED column that means the frequency of oscillator must equal or greater $10 \times 25 = 250\text{Hz}$.

b. Counter & decoder:

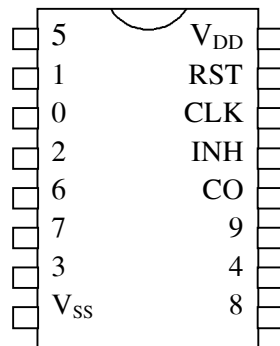
There are many kinds of counters, in this design by requirement for 10 outputs. We can use IC BCD counter and from BCD outputs we put them into IC BCD decoder to make separate outputs but for compact purpose we use IC has counter and decoder function. This IC just needs clock pulse for input and make outputs. That is IC 4017. 4017 is monolithic IC, it was produced by CMOS technology, with 16 pins. Inside IC has 5 stage Johnson counter, IC is used to count pulse for 10 radix.

Specification of 4017:

- Working voltage 3 ÷ 18V.
- Working frequency up to 12MHz.

- Input gate uses NAND trigger gate so no need to standardize input pulse.

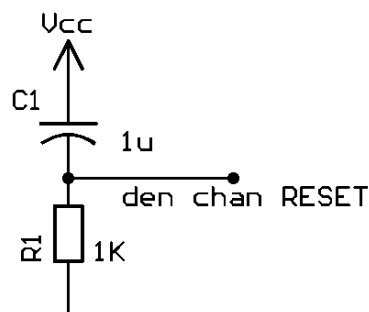
Pin assignment:



Function of each pin:

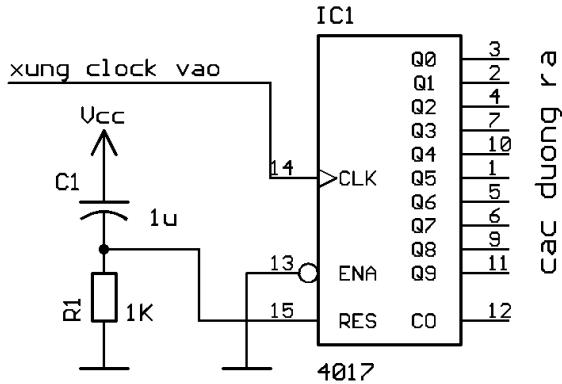
- Pin 14: input clock signal, this pulse comes to NAND trigger gate, so no requirement for too steep roll-off pulse, that means no need to make square trigger pulse by using external circuit.
- Pin 13: input of an inverting gate, output of this inverter connect to an input of a NAND gate. So when this pin is at high level, it will put a input of NAND to low level, this makes NAND gate locked and blocks all clock pulse to pin 14. At normal operation, this pin is connected to mass.
- Pin 15: input of a inverter gate and output from this gate connects to R\ pins (pins to reset FF stages). So when pin 15 at high level, it will make low logic level on R\ pins of FFs and make reset. Normally, this pin must be at low level or connect to mass.
- Outputs by order 3, 2, 4, 7, 10, 1, 5, 6, 9, 11. All outputs have buffers so output current can up to 10mA. These outputs are high active.
- Pin 12: overflow number.

At the time supply power, IC needs RESET, so to obtain RESET automatically, we need AUTO-RESET circuit as follow:



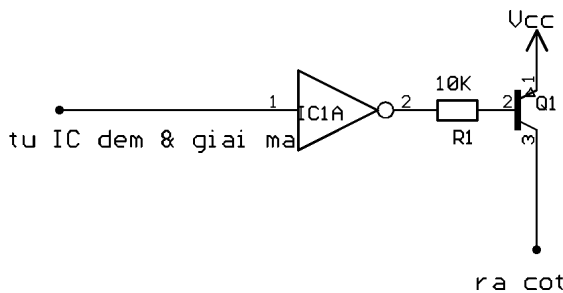
Time:
 $T = RC.$

Schematic diagram of 4017 as follow:



Outputs are connected to electronics switches and simultaneously connected to drivers to sweep columns.

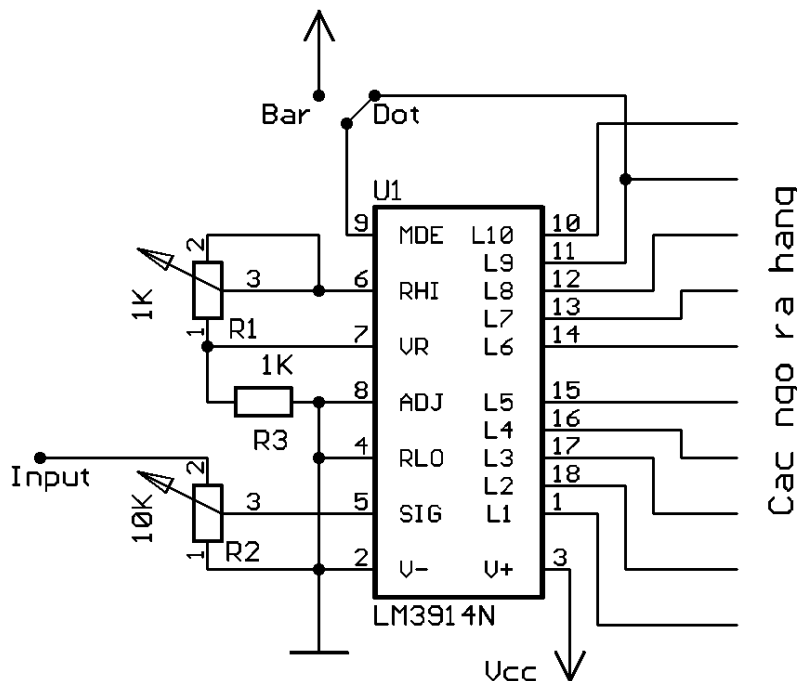
c. Driving circuit:



Drivers have function to increase output current to drive LEDs. Supply current from Vcc for LED columns. Inverters is used to isolate with previous stage and invert data from counter & decoder.

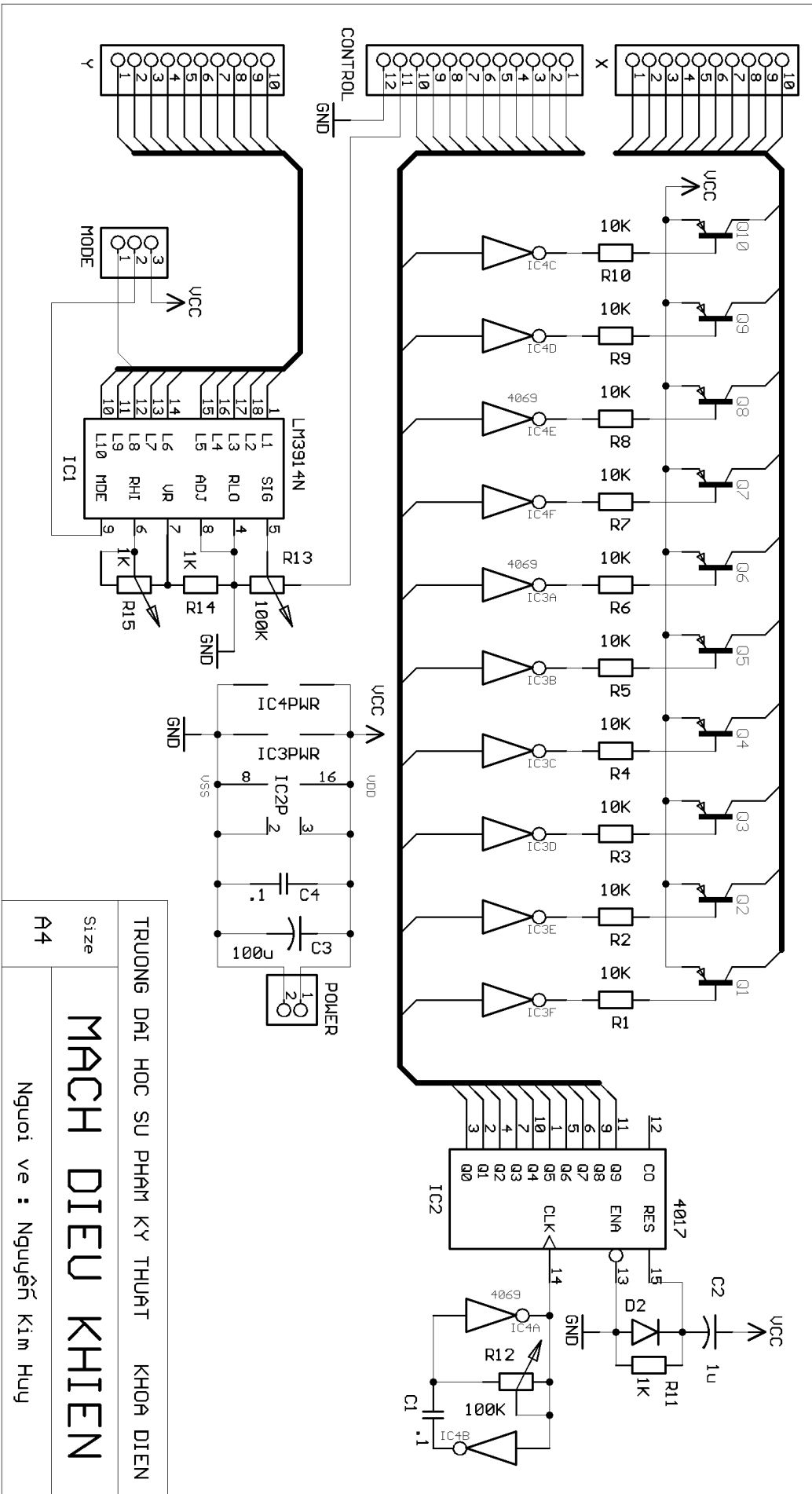
d. VU-LED circuit:

VU-LED circuit uses chip LM3914 as above mentioned part.



Following is complete schematic diagram of oscillator, counter & decoder, and VU-LED circuit:

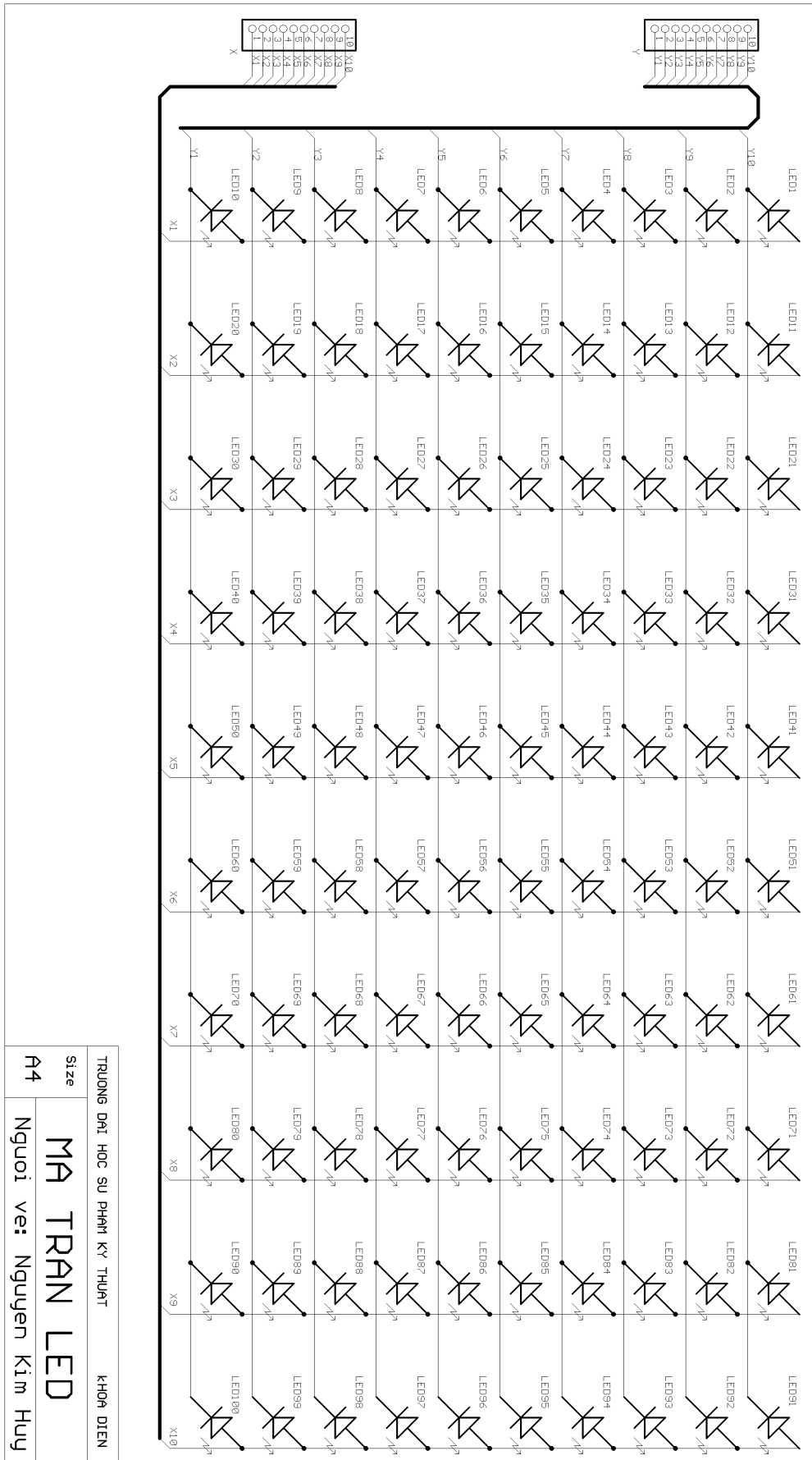
Schematic diagram of oscillator, counter & decoder circuit and VU-LED circuit



TRUONG DAI HOC SU PHAM KY THUAT		KHOA DIEN	
Size		MACH DIEU KHIEN	
A4		Nguoi ve : Nguyễn Kim Huy	

3. LED display matrix:

LED matrix is constructed by 10 rows & 10 columns. With schematic diagram as below:



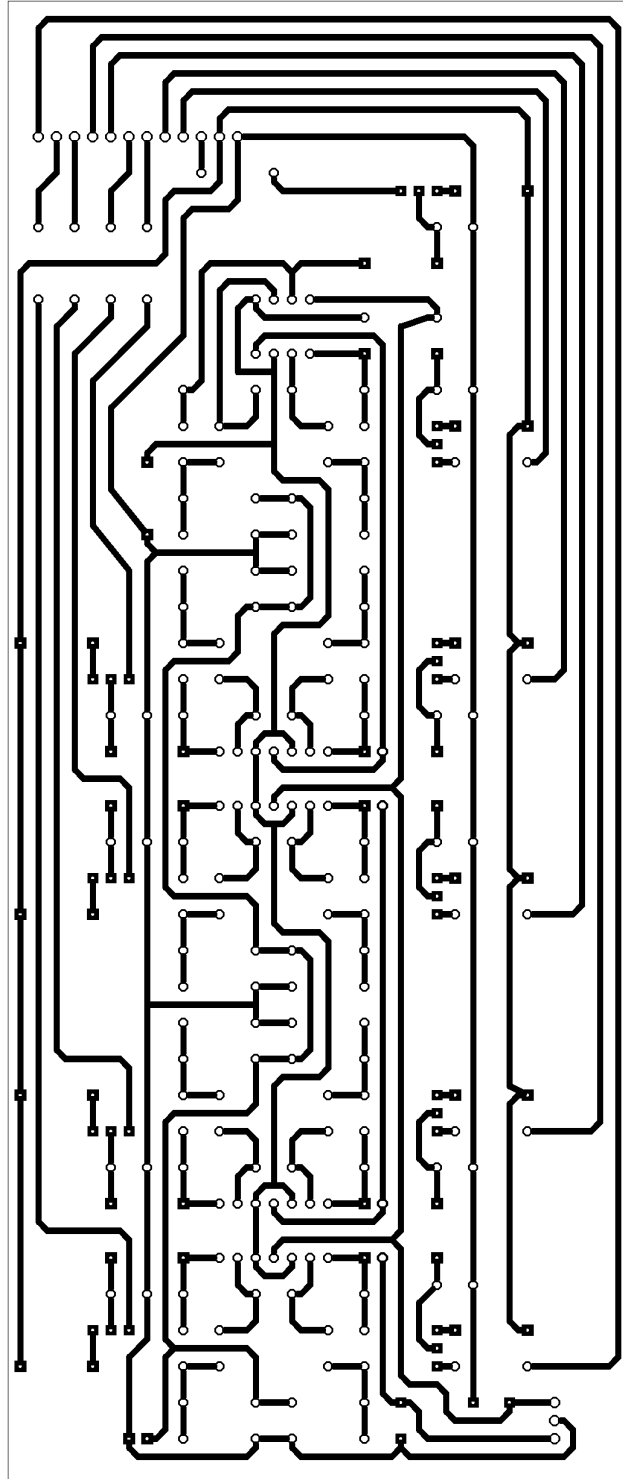
Schematic diagram of LED Matrix

TRUONG DAI HOC SU PHAN KY THUAT		KHOA DIEN	
Size	MA TRAN LED		
A4	Nguoi ve: Nguyen Kim Huy		

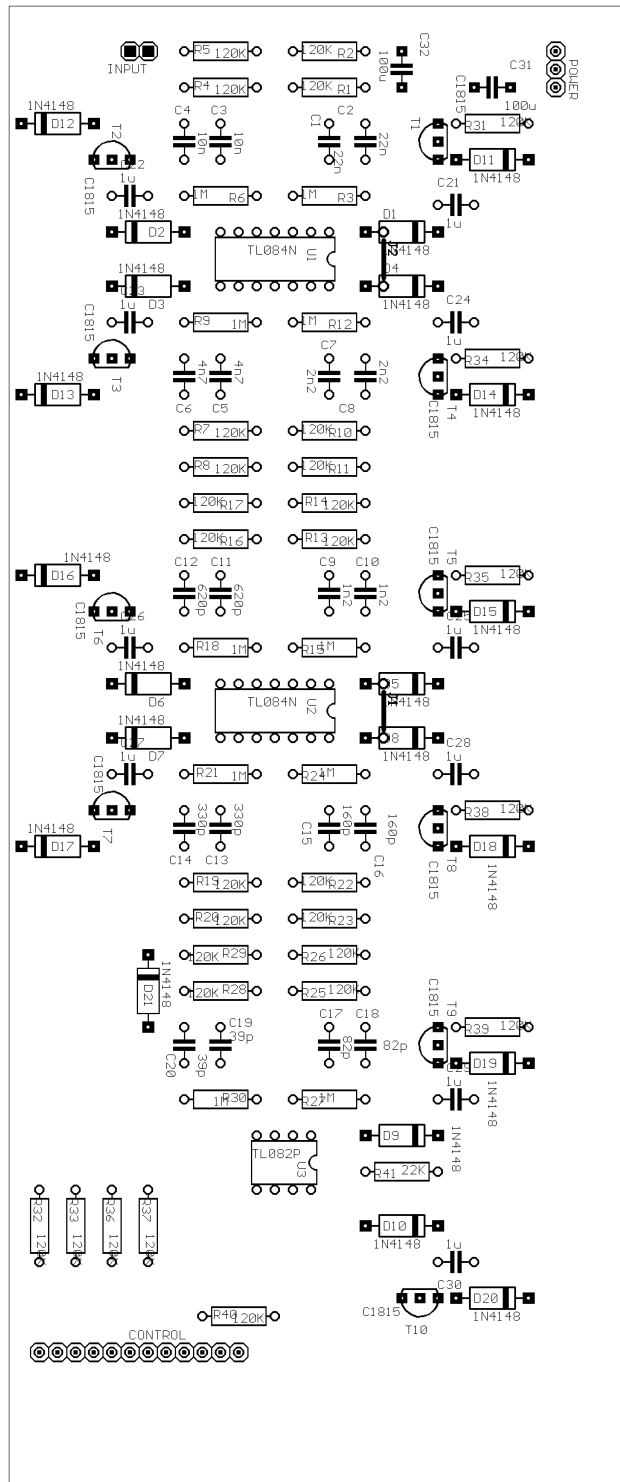
III. Design PCB:

1. Filters & switching:

PCB

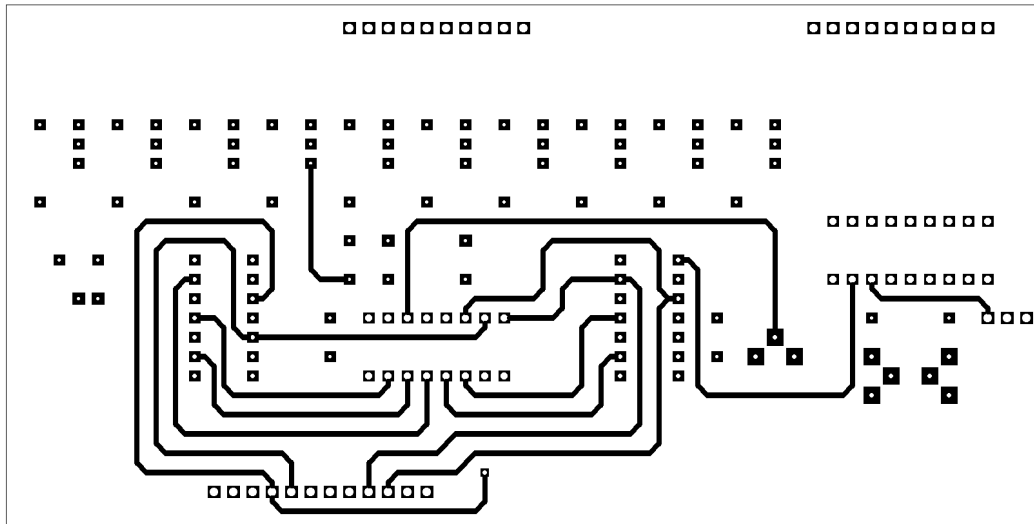


Components view

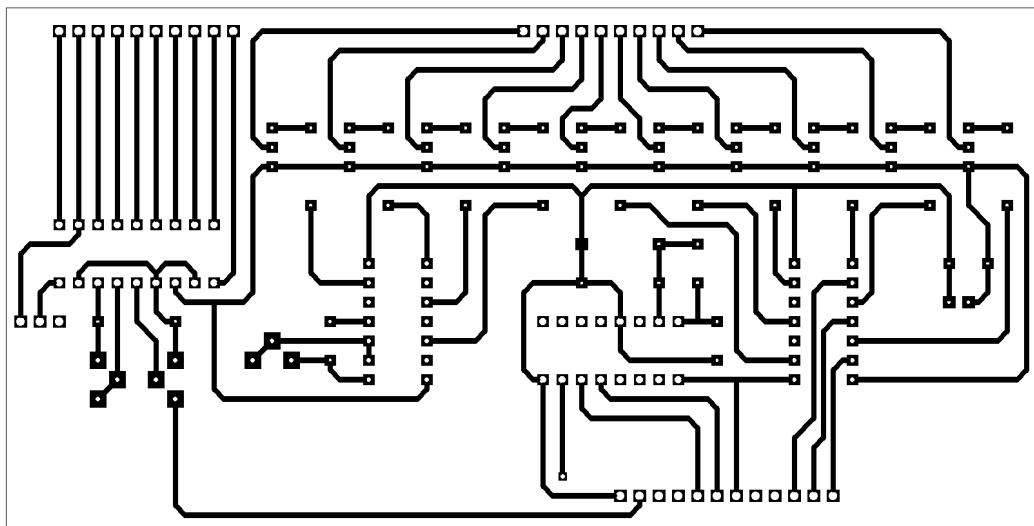


2. Oscillator, counter & decoder, and VU-LED:

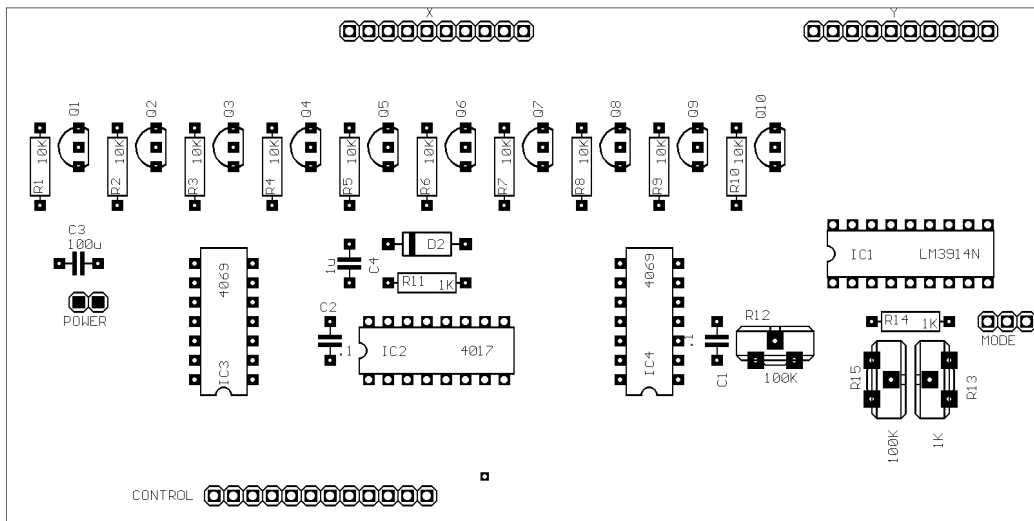
Top side



Bottom side

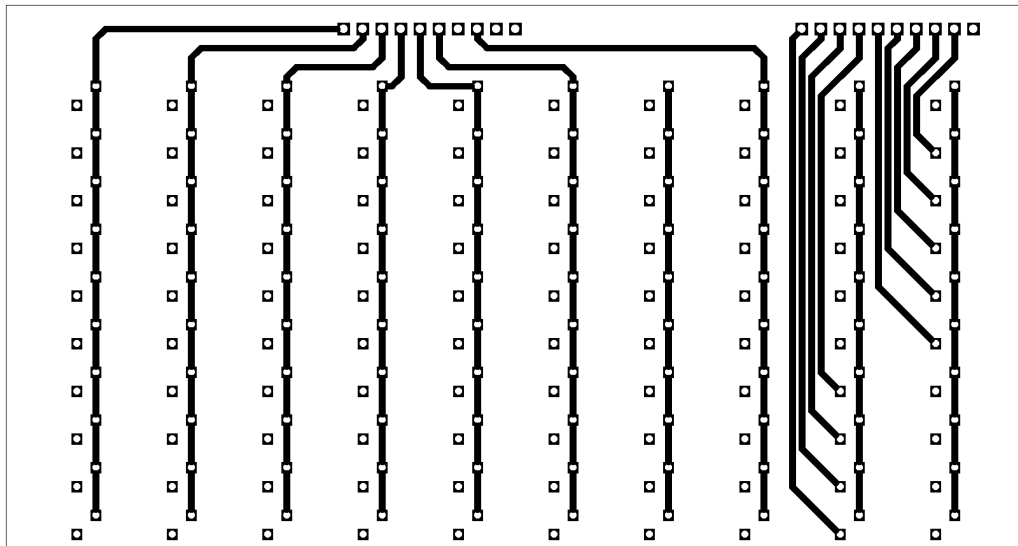


Components view

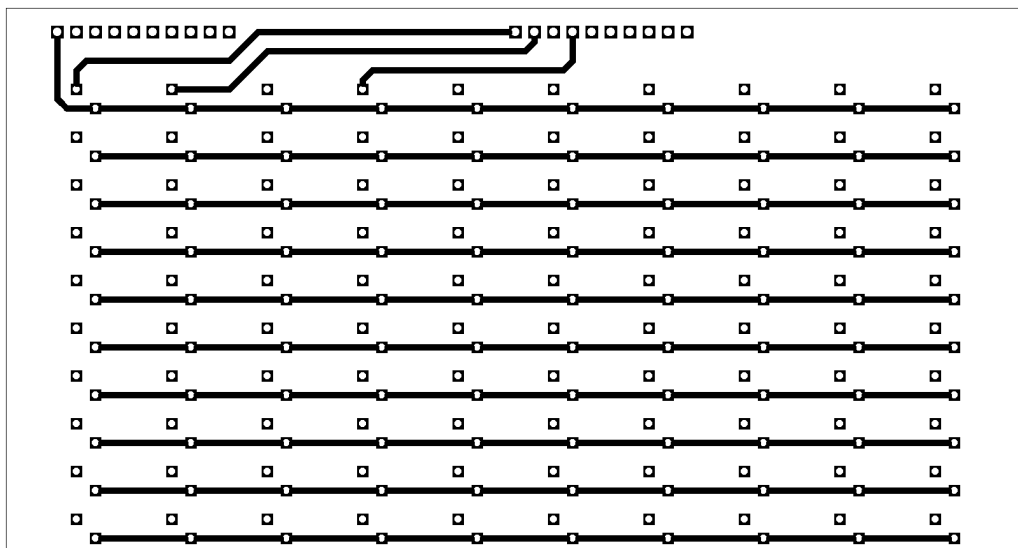


3. LED Matrix:

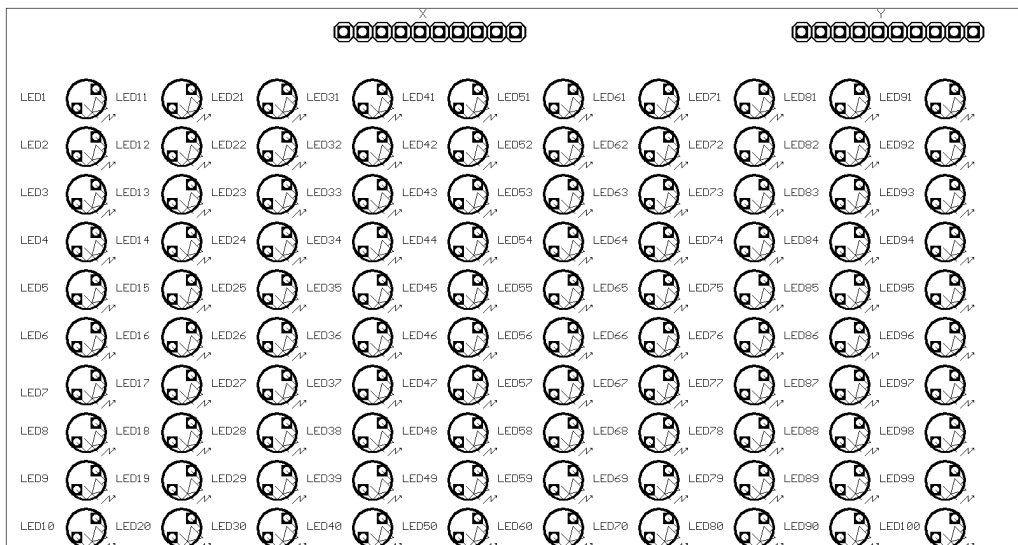
Top side



Bottom side



Components View



Conclusion

The audio spectrum analyzer was made and operated stably, meets requirements for professional audio spectrum analyzer. However, display is not showed the sense of beauty because using LEDs, if we use LCD display matrix, it will make better result.

If we combine the audio spectrum analyzer with a tone equalizers, we will have a Graphic Equalizer. This equipment is used commonly in medium rank to high rank audio systems. The combination of the two circuits is also the development for this project and finally the author hopes this is a useful document for who interesting in this project.